Design of Pulse Code Modulation (PCM) in Digital Communication using MATLAB Simulink

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Abstract: A study is carried out on the design of Pulse Code modulation (PCM) as one of the digital source coding techniques of Digital Communication. The main objective of the effective Digital Communication process is to reduce the bandwidth. The Source Coding Technique is a type of Analog to Digital Conversion. In this paper, I have focussed on the processes like sampling, Quantization and encoding. The types of sampling, quantization and encoding techniques are discussed in detail. For efficiently representing the information in the binary format after encoding from the analog source and also reducing the size of data by reducing the no of bits, Pulse Code Modulation is used. The main objective is to reduce the noise by transmitting the digitized signal. Pulse Code Modulation is designed using MATLAB Simulink.

***IndexTerms* – Sampling, Quantization, Encoding, Pulse Code Modulation, Regenerative Repeater, Companding.**

1.INTRODUCTION

Digital Communication is the mode of communication which occurs when the information is encoded digitally as discreet signals (1 or 0) & then is electrically transmitted to the recipients. [11] In the design of large and complex designs, it is often necessary to have one device communicate digital in and from other devices. One advantage of digital information is that it tends to be far more resistible to transmitted & interpreted errors than information symbolized in an analog medium. Other advantages of digital communication are Easy multiplexing, Easy processing & system performance monitoring (QOS), Regeneration of Signal, Operation at low SNR (Signal-to-Noise-Ratio), Integration of transmission & switching easily and also Voice & Data integration. However, there are certain drawbacks of Digital Communication including increased bandwidth, Higher Complexity and performance degradation due to use of ADC and DAC. Digital Communication system is used in military applications for safe communication and missile guide. It is also used in image processing for pattern recognition, robotic vision and image enhancement. [8] It is also used in digital signal processing. [1]

Source coding is the process of decreasing the number of redundant bits of information to reduce bandwidth. Reducing number of redundant bits is done by removing the unimportant extra data which is conveyed to the receiver. It encodes the analog source data into the binary format. It also reduces the size of digital source data. It contains the process of Sampling, Quantization & Encoding. [2]

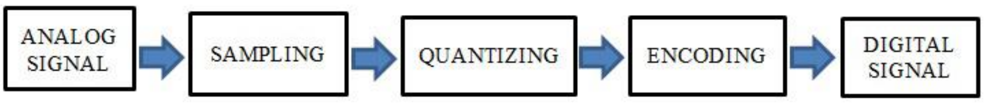

Figure 1: Transmitter Section of Source Coding

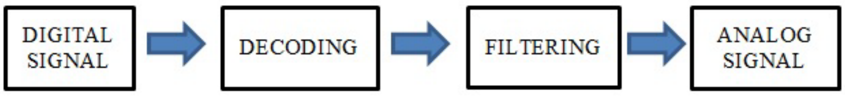

Figure 2: Receiver Section of Source Coding

2. Sampling

A Sampling is defined as, "The process of measuring the instantaneous values of continuous-time signal in a discrete form." [9] Sample is a piece of data taken from the whole data which is continuous in the time domain. When a source generates an analog signal and if that has to be digitized, having 0s and 1s i.e., Low or High, the signal has to be discretized in time. This discretization of analog signal is called as Sampling.

The process of Sampling is governed by the Nyquist Sampling theorem which states that "A bandlimited signal can be reconstructed exactly if it is sampled at a rate atleast twice the maximum frequency component in it. [3] [10]

Mathematically, **$fs >= 2 fm$** where,
fs= Sampling Frequency
fm= Modulating Frequency

Aliasing is a phenomenon where the high frequency components of the sampled signal interface with each other because of inadequate sampling fs < 2 fm. Aliasing leads to distortion in recovered signal. This is the reason why sampling frequency should be atleast twice the bandwidth of the signal.
Practically, Signals are oversampled, where fs is significantly higher than Nyquist rate to avoid aliasing.

Sampling Techniques are of three types-
(a) Impulse Sampling
Impulse Sampling can be performed by multiplying analog input signal x(t) by impulse train of period 'T'.
Sampling Output, y (t) = x (t) * Impulse Train.
(b) Natural Sampling
Natural Sampling is similar to impulse sampling, except the impulse train is replaced by pulse train of period `T', i.e. you multiply input signal x(t) to pulse train.
Sampling Output, y (t) = x (t) * Pulse Train

(c) Flat Top Sampling
During transmission, noise is introduced at the top of the transmission pulse which can easily be removed if pulse is in the form of Flat top, i.e. top of pulse are at and have constant amplitude. Hence, it is referred as at top Sampling or Practical Sampling.

3. Pulse Amplitude Modulation (PAM)

Pulse Amplitude Modulation (PAM) is an analog modulating scheme in which the amplitude of the pulse carrier varies proportional to the instantaneous amplitude of the message signal. The pulse amplitude modulated signal, will follow the amplitude of the original signal, as the signal traces out the path of the whole wave. [4]

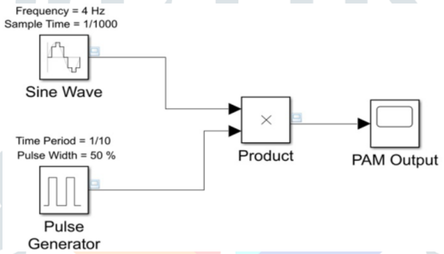

Figure 3: Pulse Amplitude Modulation Simulink Model

The following output will be there if the Sample Time Period of Pulse Generator: Square Wave = 1/10.

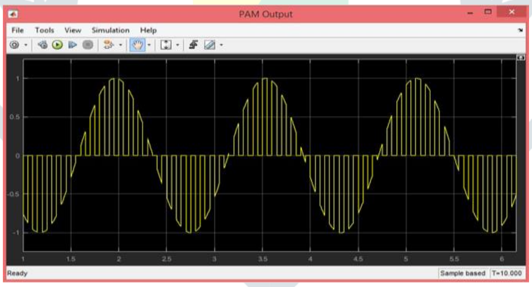

Figure 4: Pulse Amplitude Modulation Simulation Output1

The following output will be there if the Sample Time Period of Pulse Generator: Square Wave = 1/5.

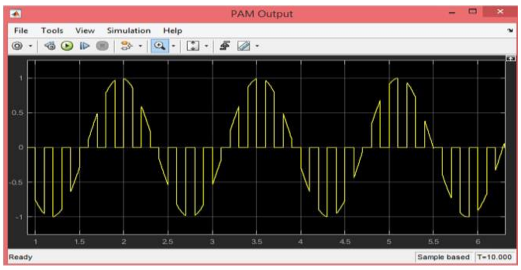

Figure 5: Pulse Amplitude Modulation Simulation Output2

The Pulse Amplitude demodulation is the type of signal demodulation (receiver) technique in which the original message signal is retrieved back from the Pulse Amplitude modulated signal using low pass filter.

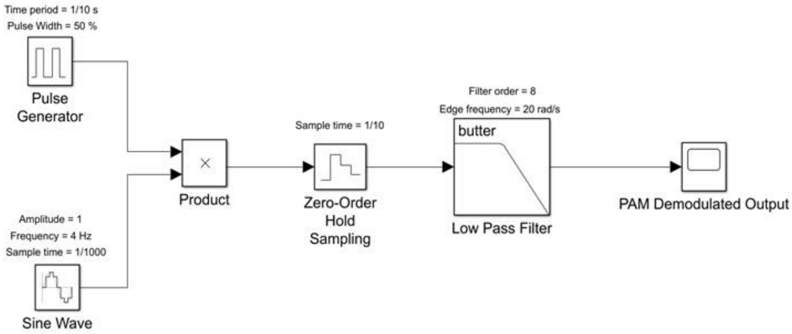

Figure 6: Pulse Amplitude Demodulation Simulation Model

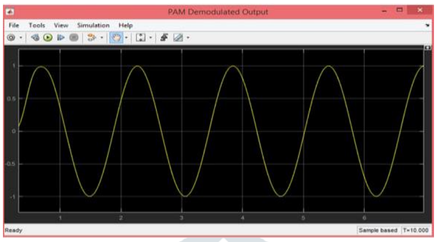

Figure 7: Pulse Amplitude Demodulation Simulation Output

4. Quantization

The digitization of analog signals involves the rounding of the values which are approximately equal to the analog values. The method of sampling chooses a few points on the analog signal and then these points are joined to round off the value to a near stabilized value. Such a process is called Quantization.

The analog-to-digital converters perform this type of function to create a series of digital values out of the given analog signal. The analog signal which is to be converted into digital has to undergo sampling and quantizing. The quantizing of an analog signal is done by discretizing the signal with a number of quantization levels. Quantization is representing the sampled values of the amplitude sample into a discrete time signal. The quality of a Quantizer output depends upon the number of quantization levels used. The discrete amplitudes of the quantized output are called as representation levels or reconstruction levels. The spacing between the two adjacent representation levels is called a quantum or step-size. [6]

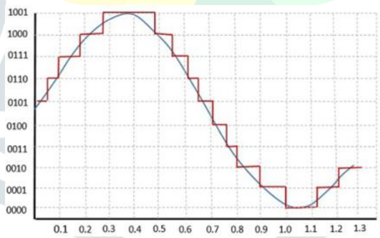

Figure 8: Quantized Signal

There are two types of quantization- Uniform Quantization and Non-Uniform quantization. The type of quantization in which the quantization levels are uniformly spaced is termed as Uniform Quantization. The type of quantization in which the quantization levels are unequal and mostly the relation between them is logarithmic, is termed as a Non-Uniform Quantization.

Both the mid-rise and mid-tread types of uniform quantizers are symmetric about the origin.

Quantization Noise is a type of quantization error, which usually occurs in analog audio signal, while quantizing it to digital. For example, in music, the signals keep changing continuously, where regularity is not found in errors. Such errors create a wideband noise called as Quantization Noise.

To overcome noise and crosstalk problems, non- uniform quantization is used as it has relatively less error. Non-uniform uniform quantization may be achieved by first passing the message through a com- pressor, non-linear device which compresses the peak amplitude. This is followed by uniform quantizer, such that uniform zones at the output correspond to the non-uniform zones at the input. [7] At the receiving end, the compressed signal is passed through an expander, another non-linear device used to cancel the non-linear effect of the compressor. This process is called COMPANDING. Two international companding standards retain up to 5 bits of precision by encoding signal data into 8 bits are u-law and A-law. U-law is the accepted standards in U.S. and Japan, while A-law is the European accepted standards.

5. Encoding

The process of assigning the binary codes to the quantizer output is known as Encoding. Encoding provides security withstand capability with the channel noise and flexible operation of signal. [3]

Encoding is of two types-
(a) Linear Encoding
In linear encoding, quantization levels are evenly spaced. It is simpler but overall signal distortion takes place in it.
(b) Non-Linear Encoding
In non-linear encoding, the quantization levels are not evenly spaced. As a result, there is reduction of overall signal distortion. It can also be done by companding and more complex as compared to linear encoding.

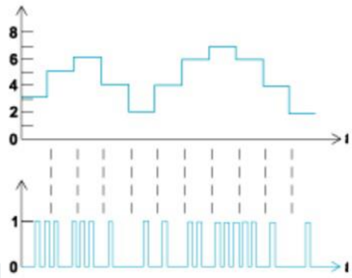

Figure 9: Quantized and Encoded Output [3]

Now these encoded data are converted into digital signal using various encoding scheme. Encoding scheme is simply mapping of data bits of signal elements. NRZ-L, RZ, AMI, Manchester, unipolar, bipolar are some examples of such encoding schemes.

6. Regenerative Repeater & Receiver

The Regenerative Repeater is used as the channel for transmission of the signal. This section increases the signal strength. The output of the channel also has one regenerative repeater circuit, to compensate the signal loss and reconstruct the signal, and also to increase its strength.

The Decoder circuit decodes the pulse coded wave- form to reproduce the original signal. This circuit acts as the demodulator.

After the digital-to-analog conversion is done by the regenerative circuit and the decoder, a low-pass filter of proper cut-off frequency is employed, called as the reconstruction filter to get back the original analog signal.

7. Pulse Code Modulation

Pulse Code modulation is the process in which analog signals are converted to digital form. The analog signal is represented by a series of pulses and non- pulses (1 or 0 respectively). The magnitude of signal is regularly sampled in uniform intervals, and then quantized in a series of binaries.

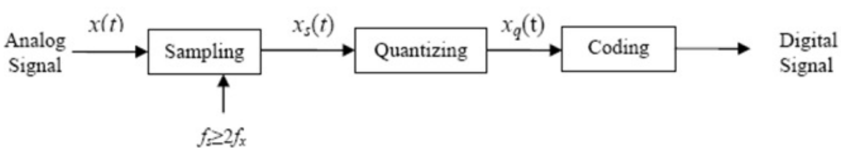

Figure 10: Theoretical Block Diagram of Pulse Code Modulation [12]

The implementation of Pulse Code Modulation is done using the three processes namely Sampling, Quantization & Encoding which I have explained above in detail. [5]

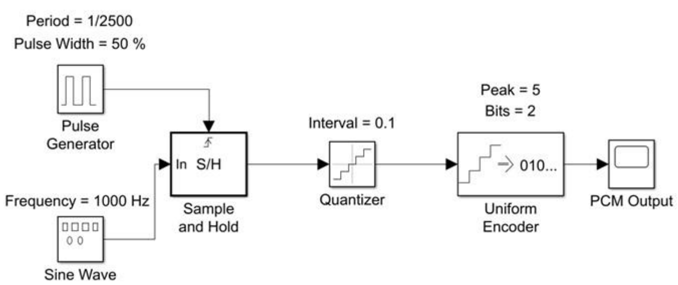

Figure 11: Pulse Code Modulation Simulink Model

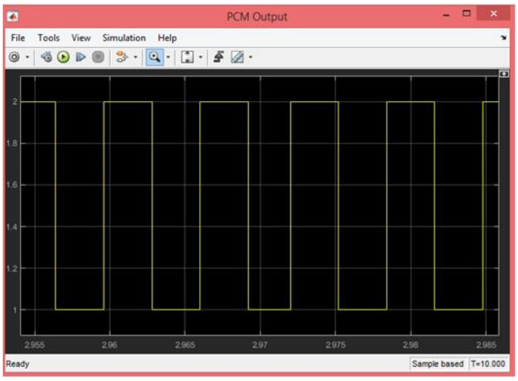

Figure 12: Pulse Code Modulation Simulation Output

The PCM Receiver circuit is used for regenerating the information signal.

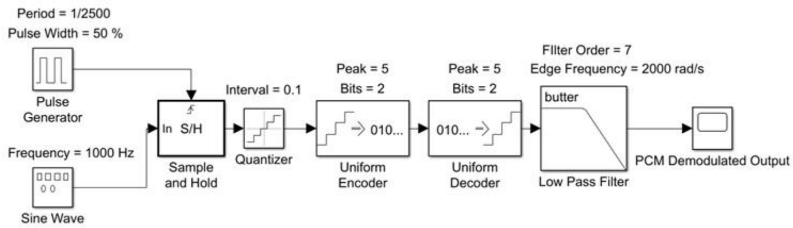

Figure 13: Pulse Code Demodulation Simulink Model

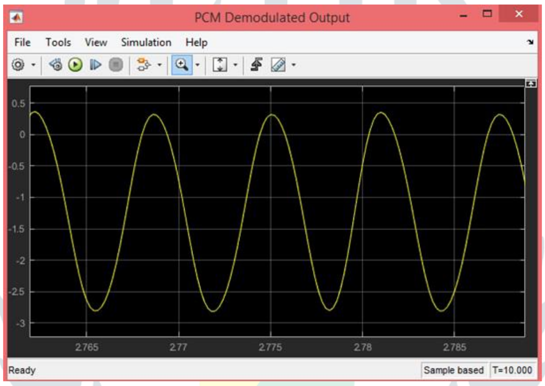

Figure 14: Pulse Code Demodulation Simulation Output

Advantages of Pulse Code Modulation Technique:
(a) PCM permits the use of pulse generation.
(b) Multiplexing of various PCM signals is possible.
(c) Due to digital nature of the signal, we can place repeaters in the transmission medium. In fact, the repeaters regenerate the received PCM signal. This can't be possible in Analog systems since amplifiers are used in it which amplifies the signal as well as noise.
(d) It has higher transmission efficiency and is convenient for long distance communication.

However, there are certain Disadvantages of PCM:
(a) It is complex, as it involves various processes like sampling, quantization, encoding and time-consuming.
(b) It requires large bandwidth since all its bits obtained after encoding needs to be transferred.

8. Conclusion & Future Scope

The Pulse Code Modulation can be implemented in CD Laser disks, digital audio recording, digitized video special effects, voice mails and also radio control. [3] These technique can be applied in the Verilog language and can also be performed in FPGA board. The Simulink design of the Pulse Code Modulation can also be done using MATLAB code and can be changed effectively on the user's requirements. To reduce the costing and complexity of the model, proper design should be studied with reduced bandwidth of signals. So, Simulink MATLAB is used to design these models easily. The major requirement is the reduction of bandwidth by proper quantization and sampling of the signal. The Simulink MATLAB is suitable for designing various physical models using various blocks included in its library.

References

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